# SPECIFICATIONS

### TOTAL HARMONIC DISTORTION
Bypass: less than .005% @ 400 Hz
Operate: less than .05% @ 400 Hz

### S + N/N
Bypass: better than -90 dB
Operate: better than -70 dB @ 10 to 20 dB G/R

### STEREO CHANNEL SEPARATION
Bypass: better than -70 dB @ 10 kHz
Operate: better than -60 dB @ 10 kHz

Gain Lock Range: -30 to -10 dB ref G/R threshold

### MULTIBAND CROSSOVER FREQUENCY RANGE
(STD , larger ranges are programmable)
Low to Mid 1 range: 80 Hz to 320 Hz
Mid 2 to High range: 2 kHz to 8 kHz
Four Band Outputs:
Level ranges: +/- 6db from reference

### GAIN REDUCTION RANGE
Wide band range: better than 40 dB
Low,M1,M2,High ranges: better than 40 dB

## MODULATION SIGNATURE SPECIFICATIONS

### INPUT (Reference 0 dBm = 0.775 VRMS)
Type: Active balanced (differential)
Impedance: > 10 K ohms balanced bridging
Termination: 600 ohms (selectable)
Level (adjustable): < - 10 dBm to +20 dBm
(Referenced to 0 dB indication on front panel level meter)

### AUX. OUTPUT
Type: Active balanced (differential)
Impedance: 100 ohms (designed to drive 600 ohm load)
Level: (Referenced to 100% modulation level as established internally)
Adjustable: < - 20 dBm to +20 dBm
Fixed: +16 dBm for connection to CRL SG-800A gen

### FREQUENCY RESPONSE
(Referenced at 20 dB below G/R, 400 Hz, +10 dBm input)
Operate Mode: 50 Hz to 15 kHz: +/- 1 dB
Proof Mode: 50 Hz to 15 kHz: +/- .5 dB

### HARMONIC DISTORTION
(+10 dBm input/output, 20 kHz bandwidth)
Operate Mode: < 0.25%, 50 Hz - 15 kHz (< 0.15% typical)
Proof Mode: < 0.1%, 50 Hz - 15 kHz

### S + N/N
(Referenced to 5 kHz at limit threshold (80% mod.) 20 kHz BW)
Operate Mode: > 80 dB, de-emphasized
Proof Mode: > 85 dB, de-emphasized

### STEREO SEPARATION (not de-emphasized, [illegible])
Operate Mode: > 60 dB, 50 Hz - 1 kHz
> 50 dB, 1 kHz - 10 kHz
Proof Mode: > 70 dB, 50 Hz - 15 kHz

### LIMITING
Adjustable from [illegible] 5 dB

### TIME CONSTANTS
Program dependent

### STEREO ENHANCE
Threshold: Adjustable from [illegible] dB
Enhance: Adjustable from [illegible]

### MODE SWITCH
Selects operate or proof mode (located on rear panel)

### FRONT PANEL INDICATORS
Dual 10 - Segment LED type input level meters with 22 dB (28 dB with OVLD) dynamic range

### LOW FREQUENCY ENHANCE
Adjustable from 60 to 90 Hz, 0 to 5 dB Boost

### PRE-EMPHASIS
Flat, 50, or 75 microseconds

### LOW PASS FILTER
Frequency Response (0 dB ref. at 400 Hz, +10 dBm input)
20 Hz to 15 kHz, +0.5/-1.0 dB

### DYNAMIC CROSSTALK PROTECTION
> 55 dB (60 dB typical)

### LIMITER AUDIO OVERSHOOT
less than 5% (0.3 dB) maximum

## STEREO MULTIPLEX GENERATOR SECTION

### COMPOSITE OUTPUT
Level: .25 volts to [illegible] volts p-p adjustable into 50 ohms
.25 volts to 9 volts into open circuit

### AUDIO FREQUENCY RESPONSE
([illegible] 0 dB @ 400 Hz, +10 dBm input)
[illegible] to 15 kHz, +/- 0.5 dB

### TOTAL HARMONIC DISTORTION
(+10 dBm input level, 15 kHz bandwidth, [illegible])
< 0.03%

### INTERMODULATION DISTORTION
(+10 dBm input level, 15 kHz bandwidth, stereo)
[illegible] 0.03%, 60 Hz/7 kHz 4:1 Ratio.

### [illegible]/N
[illegible] dB - below 100 % modulation @ 400 Hz, 75 μSec de-emphasis

### LINEAR CROSSTALK AND STEREO SEPARATION
L to R, R to L [illegible] dB, 20 [illegible] 15 kHz (55 dB typical)
[illegible] dB, 100 [illegible] to 10 kHz (60 dB typical)
L+R (Main) to L-R (Sub) > 50 dB, 50 Hz to 15 kHz (50 dB typical)
L-R (Sub) to L+R (Main) > 50 dB, 50 Hz to 15 kHz (50 dB typical)

### NON-LINEAR CROSSTALK
L+R (Main) to L-R (Sub) > 60 dB, 20 Hz to 15 kHz (70 dB typical)
L-R (Sub) to L+R (Main) > 60 dB, 20 Hz to 15 kHz (70 dB typical)

### DYNAMIC STEREO SEPARATION
> 50 dB, 50 Hz to 15 kHz

### PILOT FREQUENCY STABILITY
+/- 1 Hz, [illegible] 32 to +122 degrees F

### PILOT PHASE
[illegible]

### [illegible] PILOT PROTECTION
[illegible]

### [illegible]
[illegible]

THE PROFESSIONAL'S CHOICE

CRL SYSTEMS®

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APHEX
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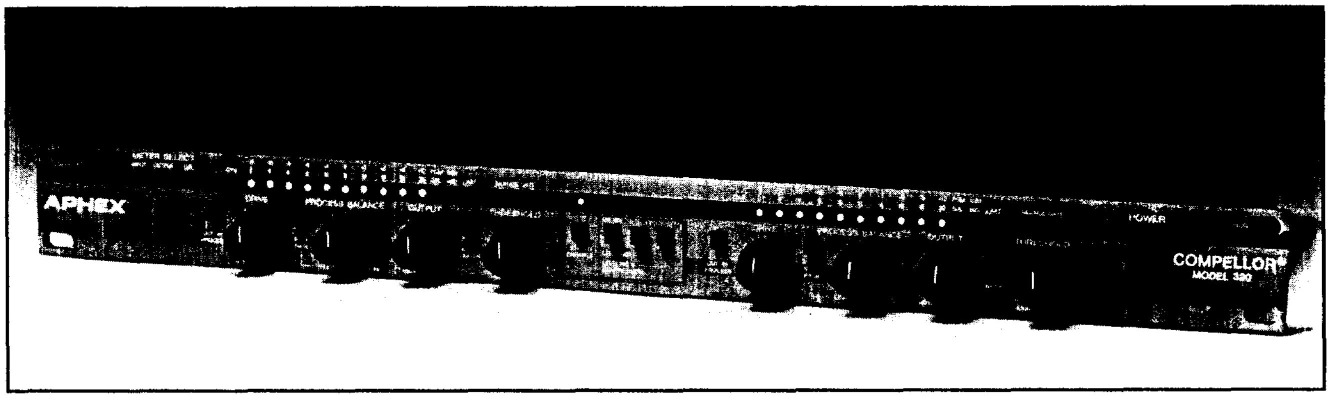
# NEW

# Compellor®
# Dual Mono/Stereo
# Compressor/Leveler
# Model 320




The Compellor Model 320 builds upon the legacy established by the Model 300 in dynamic audio processing. The 320 delivers intelligent compression, leveling and peak limiting simultaneously. Patented control circuits include analog computers that constantly analyze the input signal and vary the control characteristics. This provides for invisible operation regardless of the dynamics of program. Simply adjust the drive level to generate the desired amount of processing, set the process balance between leveling and compression and adjust the output level for unity gain. The Compellor will then provide complete dynamic control - smooth, inaudible compression, increased loudness, desired program density, and the freedom from constant gain riding - all automatically. Its unique circuitry actually enhances transient qualities, making even heavy processing undetectable.

This smart, versatile and cost effective processor is equally at home in broadcast processing, microphone control, audio recording and production, tape duplicating, live sound and film dubbing.

The 320 features dual mono operation which allows completely independent processing of two mono sources as is sometimes necessary in music recording, post production or sound reinforcement. Two modes of stereo are offered by linking the leveling control signals or linking both the compression and leveling signals. A simple metering select alternates the display of input, output or gain reduction levels. All potentiometers are detented for accurate resetting of controls. Leveling speed (fast/slow) is switch selectable from the front panel as is a defeat for the peak limiter. On the back panel, the operating reference level is switchable from -10,+4 or +8 dBm and RJ-11 connectors facilitate remote, relay bypass of the unit.

Intelligent AGC for consistent program levels

"Invisible" compression for tighter dynamics without audible effect

Instantaneous peak limiting for equipment protection (with Defeat)

Adaptive control circuits for simple set-up and no readjustment for varying programs

# *Aphex Compellor Model 320*
# *AUDIO COMPRESSOR/LEVELER/PEAK LIMITER*

The COMPELLOR's simple audio path is composed of a servo-balanced input stage, the world renown Aphex 1001 VCA, and a new, electronically, servo-balanced output stage which can be used balanced or single-ended. The nominal operating level of the COMPELLOR (and 0VU on the meter) is rear panel selectable for -10, +4, and +8dBm to match any system.

There are three main detector circuits for compression, leveling and peak limiting.

**LEVELING** is performed in a manner related to the way the ear perceives loudness over long time intervals. The circuit maintains output level within 1dB for a 20dB input level change. The action is slow enough to have minimal effect on program transients or short term dynamics.

When leveling and compression are used together, the leveler maintains the gain platform so that compression is consistent over varying levels of material, providing a uniquely smooth sounding dynamic compression.

The leveling action is interactive between the two channels when the leveling link button is depressed, one control signal is used to preserve overall balance and stereo imaging.

**COMPRESSION** is also accomplished over a 20dB range of input levels, with the ratio varying from 1.1:1 to 8:1, the attack and release times derived from, and varying with, the program material. This "soft knee" helps to prevent the "choked" sound usually associated with deep compression. Further program dependent characteristics are imparted by other sections of the COMPELLOR's computer, the DYNAMIC VERIFICATION GATE™ (DVG), and the DYNAMIC RECOVERY COMPUTER™ (DRC).

The **DVG** monitors short term and long term average levels, compares them, and impedes gain changes when program dynamics might be sacrificed for arbitrary gain reduction. The DVG also prevents gain release during short term program pauses which otherwise would cause "pumping" or "breathing" effects. Vocal material is especially benefited by this feature, sounding natural even when extremely compressed. DVG action is indicated by a front panel LED.

The **DRC** allows very rapid recovery from gain reduction under certain complex wave conditions. Signals that are high in peak amplitude but low in relative power can cause an increase in compression release rate. Unrequired gain reduction is thus inhibited, preventing loss of transient wavefronts, holes, etc. The sonic benefit is substantial, contributing toward natural, open sound, even when highly compressed.

The **PEAK LIMITER** provides further dynamic control, holding an absolute ceiling 12dB above the nominal (0VU) level. It may be bypassed using a front panel switch.

The **SILENCE GATE** detects significant gaps in program material and freezes the processing, preventing noise "swell" or buildup common in other AGC devices, then instantly releases when program resumes.

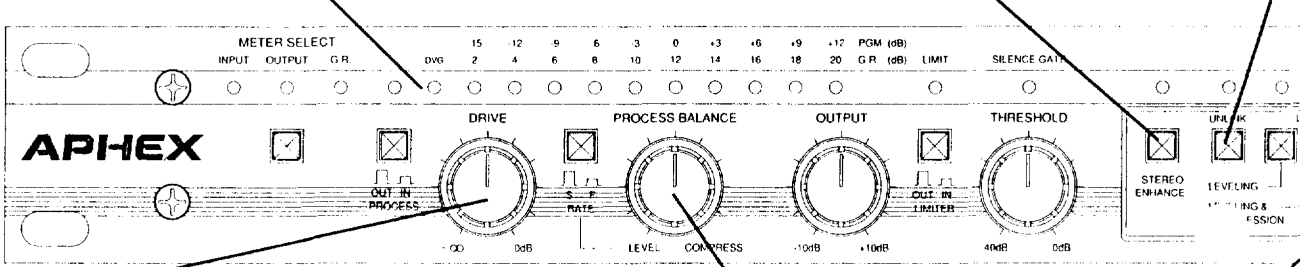
The **STEREO ENHANCE** feature does just that. By detecting and matrixing certain stereo information and sending it to the sidechains, STEREO ENHANCE creates a subtle natural widening of the stereo image that is fully mono compatible. It is not a "stereo synthesizer" and it has no effect on mono or center channel material.

COMPELLOR, Dynamic Verification Gate and Dynamic Recovery Computer are trademarks of Aphex Systems Ltd.

**INFORMATIVE METERING**
In the **PROGRAM MODE**, VU (average) level is shown as a red bar; simultaneously peak level is shown as a green bar above the red! This novel visual presentation of dynamic range can be switched to read input or output, allowing an instant display of changes in peak to average ratio. In the **GAIN REDUCTION** mode, the meters display compression as a green bar and leveling as a red dot on the same scale, thus showing total gain reduction at a glance.

**STEREO ENHANCE** switches in a unique detection and matrixing circuit which causes a pleasant widening of the stereo image without affecting non-stereo information. An LED indicates circuit operation.

U




**DRIVE** is a DC control that varies the output of the VCA and, thus, the amount of processing. Maximum compression and/or leveling is achieved with the control fully clockwise.

**PROCESS BALANCE** sets the ratio between compression and leveling depending on the need. A 50/50 balance is most useful, as the leveling keeps the compression constant over varying program levels.

**IN/OUT** instan
A/B comparisc
is also a failsa
LED indicates
each channel

## APPLICATIONS

### BROADCASTING

In the race for loudness it is quality which usually suffers. When required to work too hard, even the best multi-band processors degrade the audio. By pre-conditioning the signal with the COMPELLOR, the following processor is fed a signal with an optimized dynamic range, thus allowing it to be operated in its "sweet spot" without concern for possible overload. Since the COMPELLOR does not degrade the audio, the total result will be cleaner sound, with equal, or greater, apparent loudness.

A different problem faces classical stations, especially with the newly expanded dynamic range of digital audio. The quieter passages get "lost" in the ambient noise floor, which may, in a moving automobile be higher by more than 30dB. The COMPELLOR can "lift" these passages without changing the dynamic and transient feel, thereby pleasing the audiophile and commuter alike.

Another benefit of the COMPELLOR in the broadcast chain is that fader settings on the console become less critical. The sound of the station will not change from the DJ who loves the sound of the meters pegging to the DJ who is afraid to make them move.

Television broadcasters are often faced with the problem of a large difference of apparent loudness between program material and commercials. With a COMPELLOR, the apparent loudness of the program can be increased, while already heavily compressed commercials go through without further processing. The net result is consistent apparent levels from program to program and from program to commercial.

### VIDEO/FILM POST PRODUCTION

Matching levels among multiple sources, within a single source and transitions between dialogue is often a job which requires more than one person to ride gain and switch sources at the appropriate times. The COMPELLOR makes the job much simpler.

### SOUND REINFORCEMENT/SOUND CONTRACTING

Feedback is one of the biggest problems in live sound. Just when the fader on a vocal input is set, the vocalist starts to sing louder. The COMPELLOR, however, can maintain maximum level before feedback.

The COMPELLOR also shines in controlling multiple sources of different levels, such as conferences. The mic levels will all be equal in approximate loudness without changing the character of each individual's voice.

Paging systems can sound louder and clearer without any overload distortion and without increasing amplification.

### STL/PHONE LINE DRIVER

Maintaining consistent drive levels while controlling peaks is just another way of describing the COMPELLOR. High modulation of the STL can be sustained without concern for overload. Audio level will be kept well above the noise floor of phone lines or STL, again without crashing anything following the COMPELLOR.

### CARTING/TAPE DUPLICATION

Different audio levels from cart to cart is an all too typical problem. With the COMPELLOR, levels can be easily maintained to assure maximum signal to noise performance without tape saturation. The COMPELLOR is especially useful in assembling tapes from several sources with varying levels onto a single tape.

### MIC PROCESSING

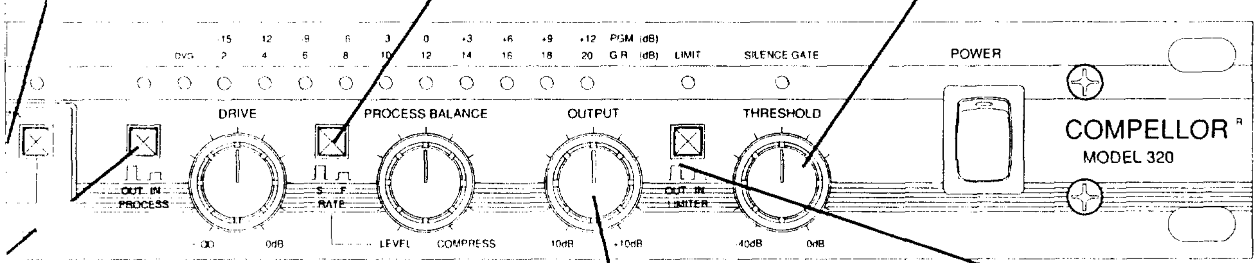
One of the most difficult signals a processor encounters is the human voice. The COMPELLOR works beautifully on voice by producing a dense, "punchy" sound while retaining dynamic and transient qualities. The apparent level will be consistent without changing the urgency and excitement of a screaming DJ or altering the intimacy of a soft-spoken female voice.

NLINK for dual mono operation.

LINK selects one of the stereo modes.: leveling control signal or compression and leveling signals linked.

**LEVELING SPEED** Fast or Slow

**SILENCE GATE THRESHOLD** sets the threshold of the SILENCE GATE between -40 to 0dB referenced to nominal input level. Below threshold the SILENCE GATE freezes gain reduction release, such as during program gaps or quiet passages. This will prevent noise buildup and permits normal fades, even with heavy processing. An LED shows SILENCE GATE action. Two SILENCE GATES for dual mono operation.

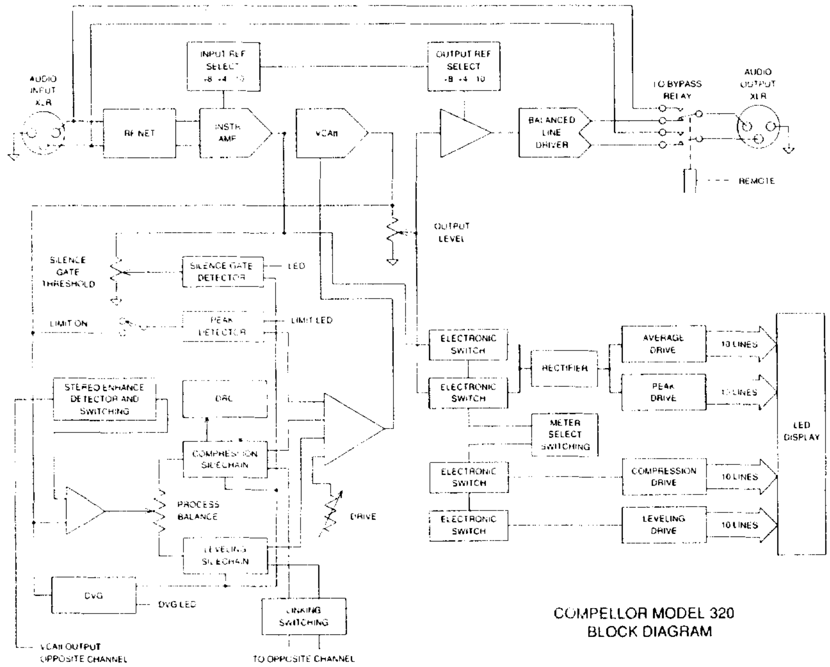



lly takes the COMPELLOR in or out of circuit for n. Sealed relays provide a hardwire bypass which e feedthrough in case of power supply failure. status as a glance (red-in, green-out). One for remote controllable), RJ-11 connectors.

**OUTPUT** adjusts the level over a 20dB range to compensate for heavy gain reduction.

**PEAK LIMITER DEFEAT**

# APHEX SYSTEMS Compellor
# Dual Mono/Stereo Compressor/Leveler
# Model 320




COMPELLOR MODEL 320
BLOCK DIAGRAM

| Specifications | (Architects and Engineers Specifications and drawings available on disk.) |
|---|---|
| **INPUT** | |
| Type | RF-filtered true instrumentation differential servo balanced |
| Input impedance | 50k Ohms balanced |
| Nominal operating level | User selectable 0VU = -10, +4, +8dBm |
| Max input level | +27dBm |
| CMRR | Greater than 60dB |
| **SIDE CHAIN** | |
| **Compression** | |
| Attack time | 5-50m Sec |
| Release time | 200m Sec-1 Sec |
| Ratio | 1.1:1-8:1 |
| Threshold | 30dB below nominal level (0VU) with drive full clockwise |
| **Leveling** | |
| Attack time | 2.5 Sec |
| Release time | 5 Sec |
| Rate | 0.5-5dB/Sec |
| Threshold | Same as Compression |
| **Peak Limiter** | |
| Attack Time | 1μSec |
| Release Time | 10m Sec |
| Threshold | 12dB above nominal level (0VU) |
| Gain reduction element | APHEX VCA 1001 Voltage Controlled Attenuator |
| **OUTPUT** | |
| Type | Electronically balanced transformerless. May be operated balanced or single-ended |
| Source impedance | 60 Ohm balanced. 30 Ohm unbalanced |
| Maximum output | +27dBm balanced or +21dBm unbalanced |
| Bandwidth | +0-1dB 5Hz-65kHz |
| Hum and noise | unity gain, +4 op level -80dBm |
| Noise refered to max output | -102dBm |
| Dynamic THD | 10dB compression, 1kHz, +4 op level, 0.05%; leveling 0.03% |
| IMD | 0.03% |
| **SIZE** | 1 3/4"H x 19"W x 10"D |
| **SHIPPING WEIGHT** | 11 lbs. |
| **POWER REQUIREMENTS** | 90-250 VAC, 50-60Hz, 20W. AC input is IEC standard receptacle, voltage select & RF filter |
| **OPERATING TEMPERATURE RANGE** | 0-40°C |
| **REMOTE LOGIC** | Short to Ground; RJ-11 Connector. X2 |


APHEX
SYSTEMS 11068 Randall Street • Sun Valley, CA 91352 • (818) 767-2929 • FAX (818) 767-2641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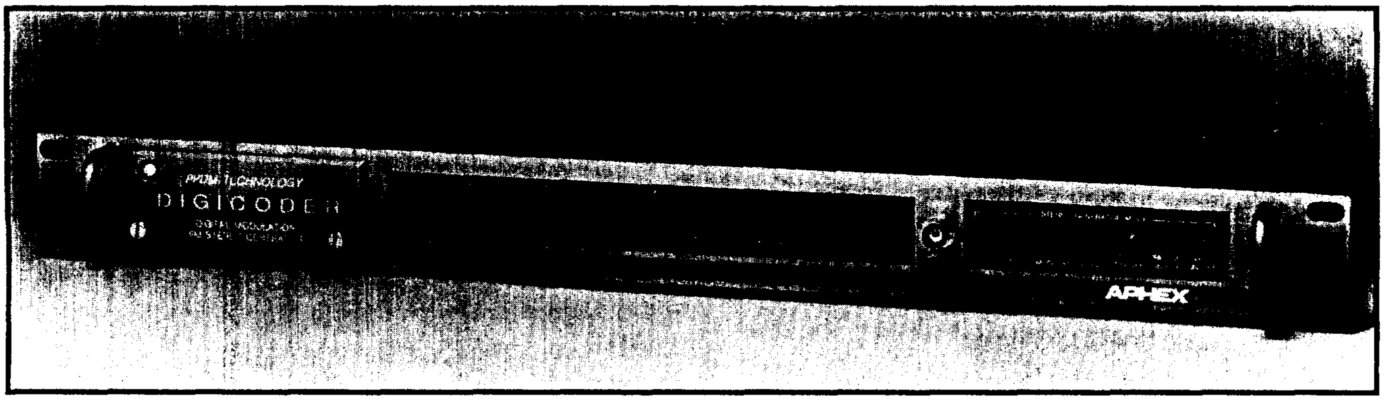
Part No. 02-320-01 Printed in U.S.A.

APHEX®
SYSTEMS

# Digicoder™ Stereo Generator

## with High Frequency Limiter, Lowpass Filter

## Model 400




**FEATURES:**

Audiophile Analog Signal Path — Audio Is Never Digitized

Digital Control — Lifetime Stability, No Internal Calibrations Necessary

Better Than 70dB Separation to 15kHz

True Real-Time Throughput — No Processing Delay

Natural Sounding High Frequency Limiter — No Voice Distortion

Zero Overshoot Lowpass Filter

High Resolution Meter

Two Independent Adjustable Transmitter Outputs

Remote Control and Tally

Quick and Easy to Set Up

# *Aphex Digicoder*
# *High Frequency Limiter, Lowpass Filter, Stereo Generator*
# *Model 400*

The sound of any FM broadcast is dependent on the quality of each piece of equipment in the broadcast chain and how well each piece of equipment is used. Unfortunately, many broadcasters use additional equipment to overcome or hide the defects of the chain rather than fix the weak links. It has been our design philosophy to determine where the weak links are and fix them. The results of that philosophy have been the world standard Aural Exciter, the Compellor, the Dominator, and now the **Aphex Digicoder**.

The stereo generator (stereo encoder) has traditionally been one of the weakest links in the chain and the one most often compensated for.

There have been several different methods used for stereo generation which have performance limitations. The introduction of the **Digicoder** eliminates these limitations and provides sonic performance far greater than any competing technology.

The **Digicoder** is a multiplex encoder for FM stereo combining the functions of pre-emphasis limiting, the 15kHz low pass filter and stereo encoding, plus RDS input and output. It uses the highest attributes of both digital and analog signal processing to produce unmatched stability, accuracy, ease of set-up and use, and sound quality.

**PANEL SECURITY SWITCH**— Switches Panel Security. When "on" the front panel Stereo Generator Mode switches are defeated and the mode can only be changed via the remote control. The Meter Function Select switches are unaffected.

**PANEL SECURITY INDICATOR LED**— LED is lit when Panel Security is switched "on".

**HARDNESS**— Adjusts ratio of sliding filter action to clipping of the limiter. CCW is more sliding filter action, less clipping (soft). CW is less sliding filter action, more clipping (hard).

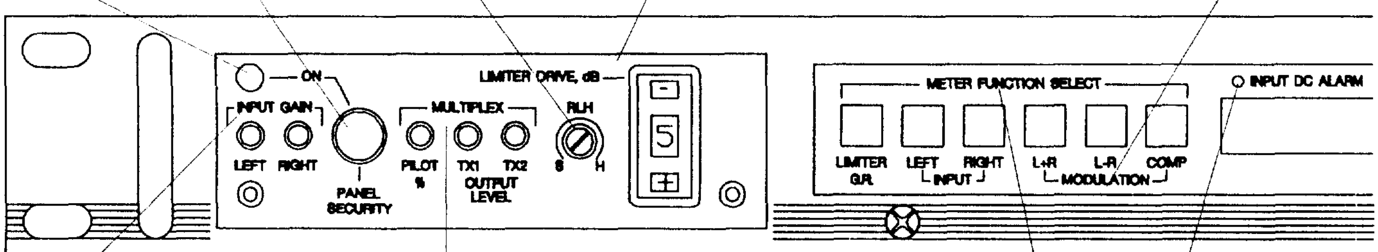
**THRESHOLD**— Adjusts and indicates the maximum amount of limiting in 1dB steps from 0 to 15dB. "-" reduces the amount of limiting and "+" increases the amount of limiting. The greater the amount of limiting the closer the output will stay relative to 100% modulation.

**MODULATION**

**L+R**— The meter sh modulation of the mo

**L-R**— The meter sho mopdulation of the st

**Composite**— The m percentage modulati composite signal.




**INPUT ATTENUATORS (LEFT, RIGHT)**— Twenty turn trimmers adjust left and right audio inputs from 1 to 10V Peak.

**MULTIPLEX**

**PILOT MIX**—Twenty turn trimmer adjusts pilot tone insertion from 5 to 13%.

**OUTPUT (TX1, TX2)**—Twenty turn trimmers adjust composite output levels for two separate outputs (TX1 and TX2) from 0 to 7V Peak.

**METER FUNCTION SELECT**

**LIMITER GR**— The meter shows the amount of action of the limiter from no indication (no limiter action) to 100% where it indicates the greatest amount of action.

**INPUT (Left, Right)**— the meter shows input relative to maximum input for 100% moudlation for left and right channels.

**ALARM EXCESSIVE** lights when more than at either input.

ıe audio signal remains pristinely in the analog ɔmain throughout so there are no "A to D" and "D A" convertors to degrade the signal. The 'e-emphasis limiter is designed to sound natural ıd unprocessed while allowing loudness to be aintained. The low pass filter has excellent əquency response in the pass band and extreme ıjection in the stop band with no overshoot or ıging. By employing an exclusive "<u>P</u>arallel <u>P</u>ath igital **<u>M</u>**odulation" **(PPDM)*** scheme of subcarrier əneration, a precision multiplex signal is ənerated which possesses separation beyond at measurable by most laboratory equipment, ‹treme stability of operating parameters and the ghest audio transparency achieved in an FM stereo encoder. The composite output is completely stable and requires no composite clipping.

The use of the **Digicoder** will allow a broadcaster to turn down the amount of processing and still maintain the same level of loudness. It is quick and easy to set up and once it is set up it <u>never requires recalibration</u>.

The total sonic result is a clean, open stereo signal with accurate, stable imaging. The broadcast will have the maximum allowable deviation with no overmodulation.

*PPDM Patent Pending

ɔws percentage
no signal.

ws percentage
əreo signal.

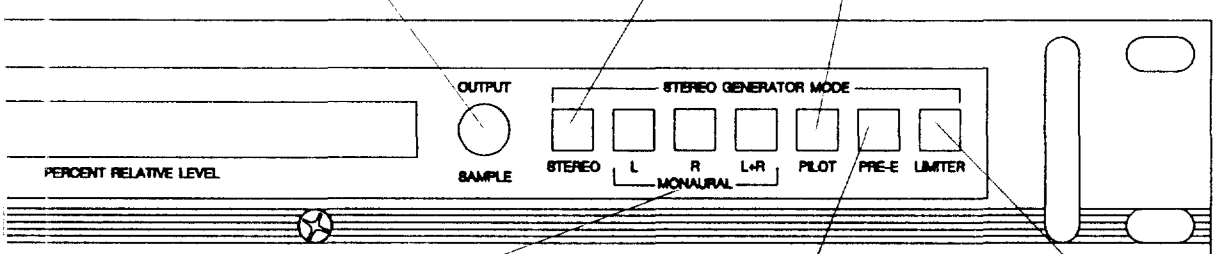
əter shows
ın of the

**SCOPE**— Standard BNC connector for monitoring composite output before any level adjustments from TX1 or TX2 trimmers.

**STEREO**— The composite will contain Left and Right channel inputs.

**PILOT ON/OFF**— switches the pilot tone mixed into the composite on. The LED is lit when the Pilot is "on".

**NOTE:** The pilot tone will **not** be mixed into the composite in any mono mode even though the light is "on".




NPUT DC— LED
5VDC is present

**MONAURAL**

**LEFT**— The composite will contain only Left channel input (Pilot off).

**RIGHT**— The composite will contain only Right channel input (Pilot off).

**Left + Right**— The composite will contain the sum of Left and Right channel inputs (Pilot off).

**PRE-EMPHASIS ON/OFF**— Switches the pre-emphasis in the inputs on. The LED is lit when the Pre-emphasis is "on".

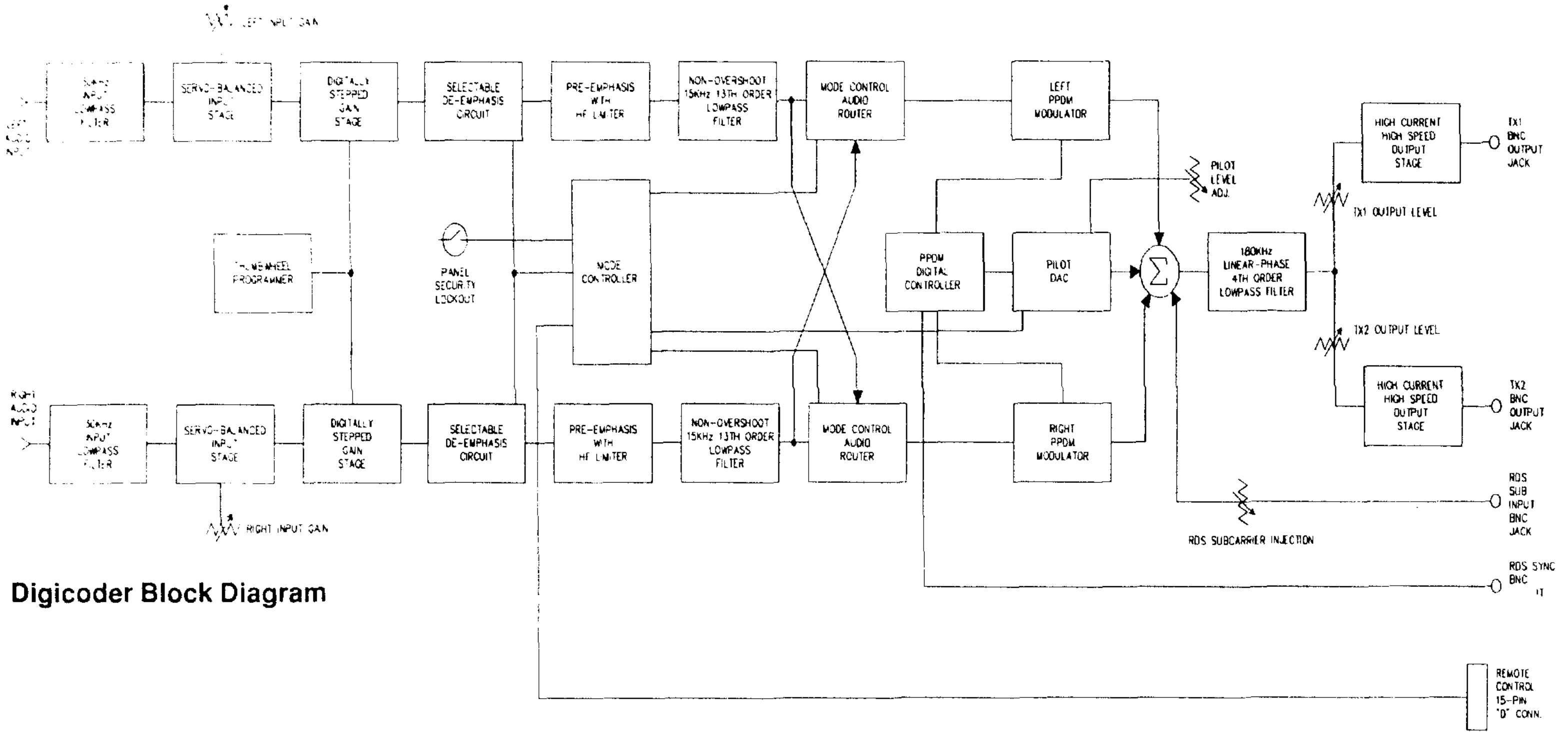
**LIMITER ON/OFF**— Switches the limiter's sliding filter action on. The LED is lit when the limiter is in circuit.

# APHEX
SYSTEMS

# *Digicoder™*
# *Stereo Generator*

## *with High Frequency Limiter, Lowpass Filter*

# *Model 400*




**Digicoder Block Diagram**

### SPECIFICATIONS

Output reference = 100% peak modulation of 100Hz sinewave in stereo mode with 9% pilot injection except where noted otherwise. Measurements are made using a properly calibrated Belar Model FMS-1 Stereo Modulation Monitor connected directly to the Digicoder TX1 or TX2 multiplex output jack or computation from spectrum analyzer indications of the composite output signal.

| | |
|---|---|
| Signal-to-Noise Ratio: | |
| Flat: | >80dB |
| 75µSec De-Emphasis: | >90dB |
| Left-to-Right and Right-to-Left Channel Separation: | >70dB |
| Main-to-Sub and Crosstalk: | >60dB |
| Sub-to-Main Crosstalk: | >60dB |
| Rejection of All Frequencies Above 53kHz: | >80dB |
| 38kHz subcarrier Rejection: | >60 dB |
| Pilot Phase Error: | 0 Degrees ±0.04deg |
| Pilot Amplitude Adjustment Range: | 5% to 13% |
| Pilot Frequency: | 19kHz ±0.1 Hz |
| THD (left or right input to multiplex output) 20Hz to 15kHz with the input adjusted to produce 50% peak total modulation and the limiter turned off: | <0.005% |
| SMPTE IMD (left or right input to multiplex output) with the input adjusted to produce 50% peak total modulation and the limiter turned off:: | <0.005% |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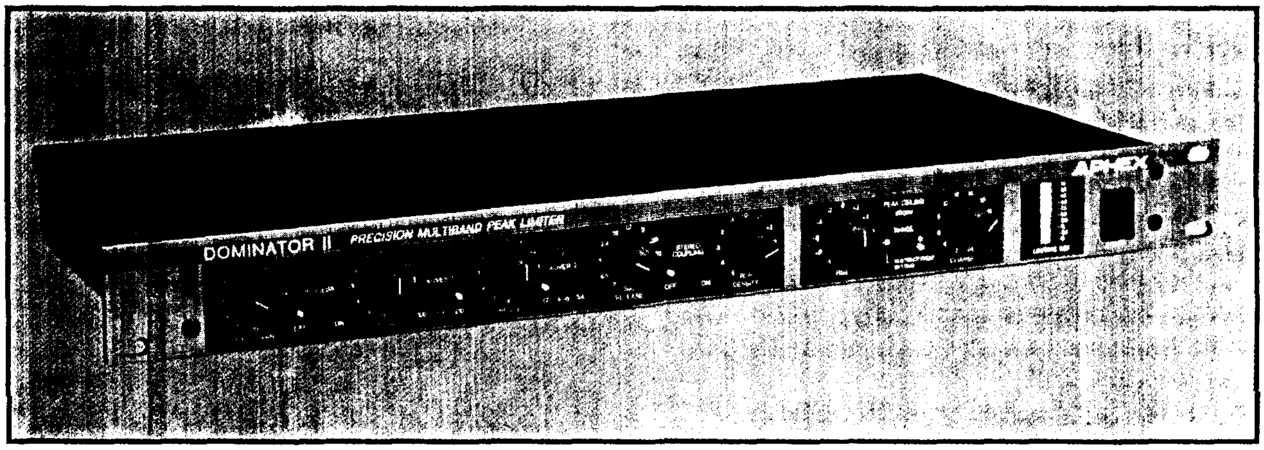
# Aphex Dominator™ II Precision MultiBand Peak Limiter Models 720, 723




The Dominator II from Aphex Systems is a stereo multiband peak limiter designed to fit a wide range of audio applications. Through the use of multiband techniques along with new proprietary circuits, the audibility of limiting action has been greatly reduced, especially when compared to conventional limiters. This means that greater limiting depth is possible, resulting in higher loudness with maintained audio quality. At virtually any limiting depth, the Dominator II is free of "hole punching", "dullness", and most other effects normally associated with limiters. As a peak overshoot protection limiter, the Dominator II is undetectable in line while it absolutely prevents peak levels from exceeding a user settable output level. In addition, the desired limiting effects of greater audio density and increased "punch" are readily available with the Dominator II.

104dB Dynamic Range

Freedom from Pumping

Freedom from Spectral Gain Intermodulation

Automatic Limit Threshold (ALT)

Peak Ceiling Trimmable in 0.2dB Steps Over a 34dB Range

Adjustable Density (Relative Crest Height)

Switchable Crossover Frequencies

Detented Potentiometers

Relay Bypass, Remote Controllable

Servo-Balanced Transformerless Inputs and Outputs

# *Aphex Dominator II*
# *Precision MultiBand Peak Limiter*
# *Models 720, 723*

## Multiband vs. Wideband Processing

A very significant problem with **wideband** processing is **"spectral gain intermodulation"** which occurs when one part of the spectrum controls the level of another part. A typical situation is a vocalist being "sucked down" every time the kick drum hits.

Since most energy is contained in the lower frequencies, they tend to control the level of the entire spectrum. When the lower frequencies are above the limit threshold the higher frequencies are attenuated thus causing the output to be dull.

**Multiband processing** solves these problems by splitting the audio into two or more frequency bands and processing each band separately. However, more bands often result in many more parameters to control including a method of summing the bands together again. While giving the user flexibility, it also requires different settings for almost every different source.

The Dominator II uses **program dependent, intelligent circuits** to reduce the number of controls. The user, therefore, has flexibility to shape the sound while quickly and easily achieving the goal of consistent, effective limiting.

## ALT (Automatic Limit Threshold)

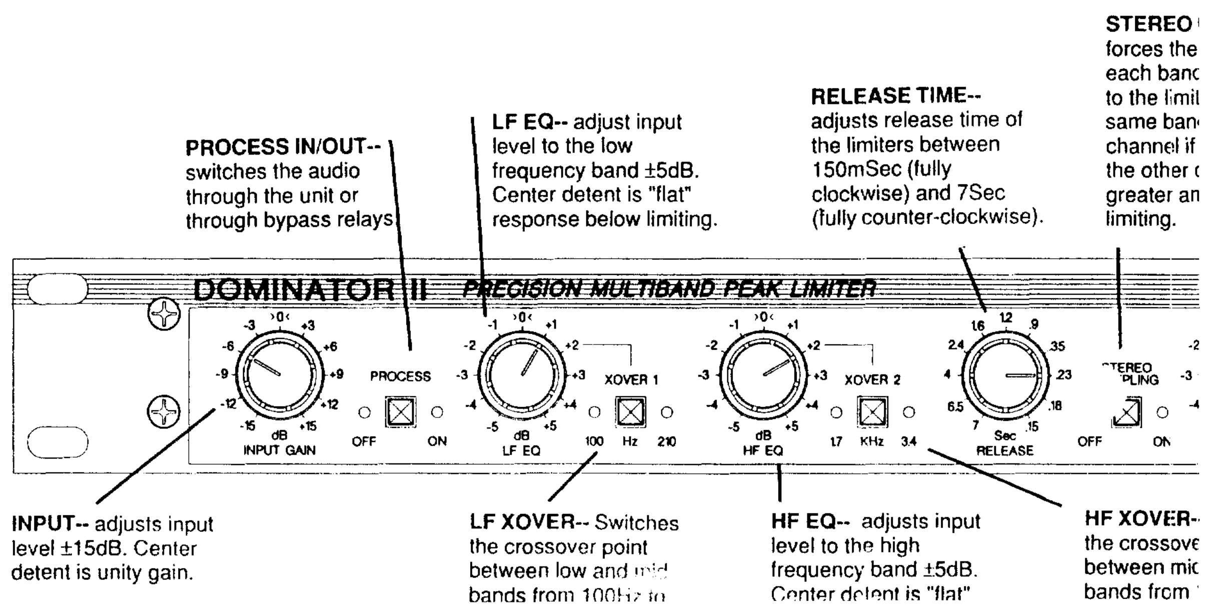
A multiband processor splits the audio into separate bands, limits each band individually and then sums the bands together again. Even though each band's peak output is predictable, summing the bands together produces an unpredictable peak output.

One conventional approach to making the summed output predictable is to use a wideband limiter after the summing. This, however, introduces all the drawbacks of wideband limiting discussed above.

Another approach is to use a clipper on the summed output. This causes too much clipping distortion if the summed output is too high. In order to avoid this distortion the limiters' thresholds are set very far below the clipper threshold. The drawback is a loss of loudness and, due to the lower thresholds, much greater amount of processing.

The Dominator II uses a patented method to produce a predictable peak output while maintaining maximum loudness without audible distortion- the **Automatic Limit Threshold (ALT)**. The outputs of the three bands are summed and sent to the ALT detector circuit. If the sum exceeds a reference value, the ALT reduces the thresholds of the individual limiters. When the summed output falls below the reference value the limit thresholds return to their original setting.

The ALT circuit has a self-adjusting finite attack time. The amount of time it takes to lower the thresholds of the limiters is the length of time the limiters' overshoot may be in the clipper. The reference value of the ALT in relation to the clipper determines the depth of clipping.



Both parameters are set by the **DENSITY** control. When the DENSITY control is set higher, the ALT reference gets closer to clipping, and the attack time is slower, producing more clipping. The opposite occurs when DENSITY is set lower. The **"0 RCH"** position for the DENSITY control emulates the standard parameters of the original Studio Dominator model 700, and is recommended for general use.

It should be noted that there is only one ALT circuit controlling both channels equally. This provides global stereo balance and imaging by assuring that both channels always limit at the same threshold. This does cause an interaction if the Dominator II is used as two independent channels. Therefore, we do not recommend such a practice.

# Model 723 Pre and De-emphasis

**Pre-emphasis** is an equalization curve expressed as a time value based on the ratio of a resistor and capacitor. The higher the value, the greater the equalization. It has been employed as a noise reduction technique for broadcast and transmission links.

There are primarily two world standards- **50 and 75 microseconds**. The Dominator II Model 723 has pre-emphasis (either 50 or 75 microsec) added after the input circuit and before the limiters. It has a complementary de-emphasis circuit (which may be switched out of circuit) after the final limiter and before the output stage.

When the **de-emphasis circuit** is in circuit the audio output of the Model 723 is flat if the input is below threshold. As the input increases above threshold the output takes the shape of the de-emphasis curve.

# Applications

**Sound Contracting** -- protection of amplifiers and speakers from overload; increased loudness; maximized use of available power.
**Recording** -- preventing sudden peak overload of mixer or recorder; tightening tracks; special effects, etc.
**Mixing** -- used as a program limiter, the Dominator II will keep a track "rock steady" for "layering" into or on top of a mix.
**Digital Sampling** -- obtaining good full scale samples free from peak overload, i.e. no more missed samples.
**Digital Recording** -- insuring clean recording by stopping clipping of peaks and overshoots. Maximizes bit usage for less distortion.
**Satellite Uplink** -- Modulation control to prevent splattering on high frequency audio, gives reduced distortion, better signal-to- noise.
**Broadcasting** -- AM and FM modulation control for increased loudness; cleaner sound; use in production for greater consistency of tapes, punchier voiceovers.
**Location Film Shoots** -- anti-crash for dialog and sound effects recording.
**Post Production** -- Soundtrack peak control; managing difficult dialog; controlling transient sound effects.
**Optical Recording & Transfer** -- prevents "valve clash", gives higher average level with low distortion and better signal-to- noise performance.
**Analog Disk Mastering** -- peak control for high allowable average cutting levels; less limiter degradation to the program; brighter, punchier sound.
**C/D Mastering** -- peal and density control for more accurate digitizing, cleaner sound requiring less error correction on playback; no limiter induced sound degradation.
**STL & Phone Line Driver** -- maximize signal-to-noise without overload distortion.
**Video and Audio Tape Duplication** -- "Hotter" transfers without saturation.

**COUPLING**--
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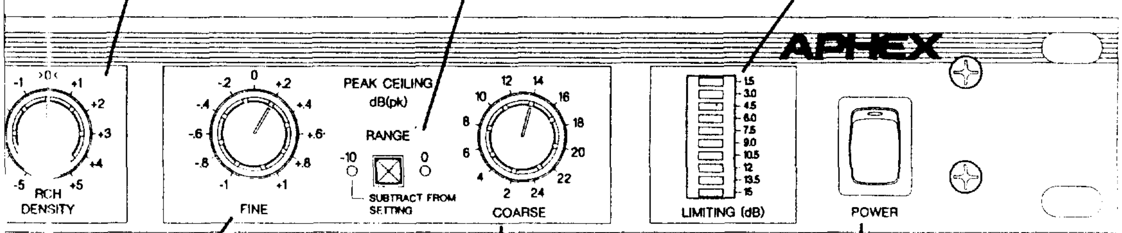
**DENSITY**-- adjusts the RELATIVE CREST HEIGHT (RCH) of the output. The higher the RCH (clockwise) the louder the output. The lower the RCH (clockwise) the lower the average level in the output.

**RANGE**-- when switched to the "-10" position it adds a 10dB boost in the input and a 10dB cut in the output. The PEAK CEILING is therefore 10dB lower than the settings shown on "COARSE" and "FINE" controls.

**METER**-- displays the limiting (from gain reduction and clipping) in the channel with the greatest amount of limiting.




· switches
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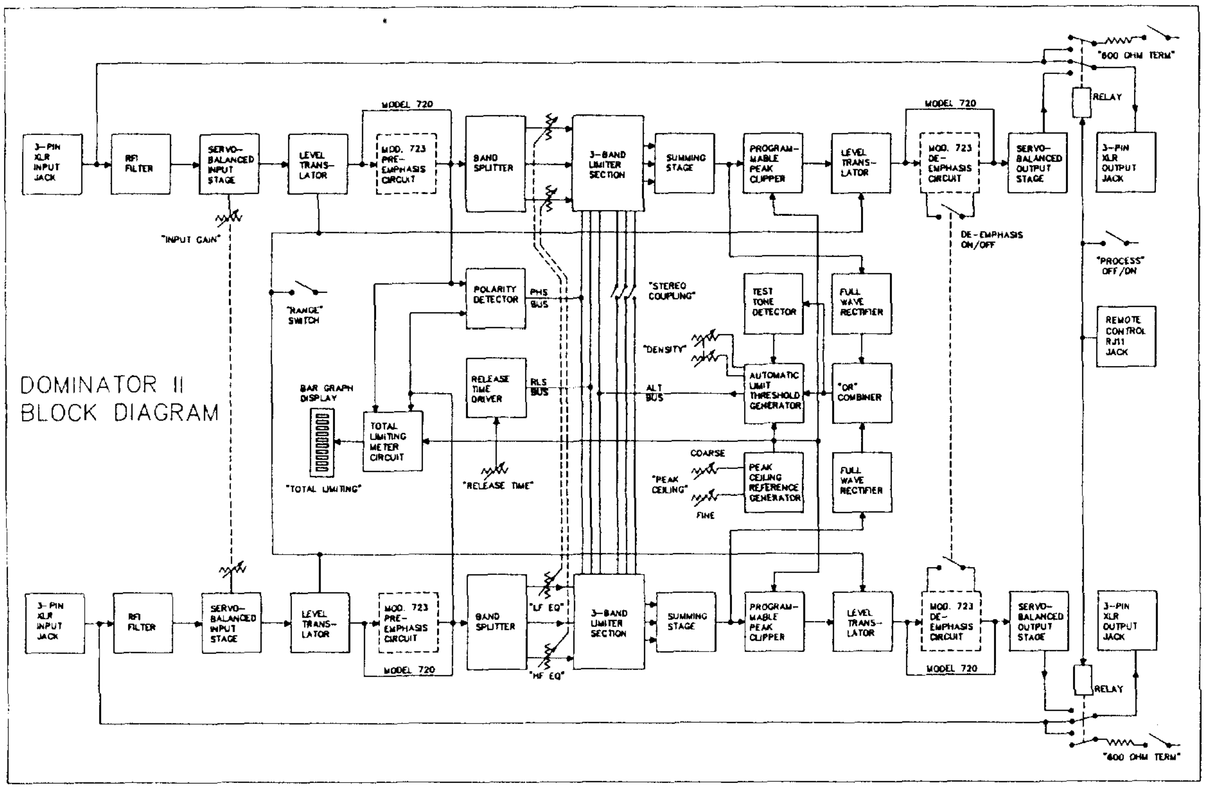
.7kHz to

**FINE**-- adjusts the PEAK CEILING by ±1dB in 0.2dB steps.

**COARSE**-- Switches the PEAK CEILING in 2dB steps from +2 to +24dB(pk)

**POWER**-- switches AC mains power on and off. "Off" position engages bypass relays.

# Aphex Dominator II *Models 720, 723*




| **AUDIO SPECIFICATIONS** | | |
|---|---|---|
| RANGE SETTING | 0dB | -10dB |
| Nominal Gain | 0dB ±15dB | Same |
| Output Noise | -81dB | -89dBu |
| THD | <0.008% | <0.008% |
| SMPTE IMD | <0.008% | <0.008% |
| DIM | <0.008% | <0.008% |
| Frequency Response | ±0.2dB 2Hz-75kHz | Same |
| Max Input (MIL) | +27dBu | +23dBu |
| Max Output (MOL) | +22dBu (RMS)* | +12dBu (RMS)* |
| Crosstalk | 70dB up to 20kHz | Same |
| Dynamic Range | 104dB | 102dB |

| **CONTROLS** | **ADJUSTMENT RANGE** |
|---|---|
| Input Gain | ±15dB |
| LF EQ | ±5dB |
| LF Crossover | 100Hz/210Hz |
| HF EQ | ±5dB |
| HF Crossover | 1.7kHz/3.4kHz |
| Release Time | 150mSec to 7Sec |
| Density | -5 to +5 RCH |
| Output Ceiling | -9 to +25dB (PK)** |

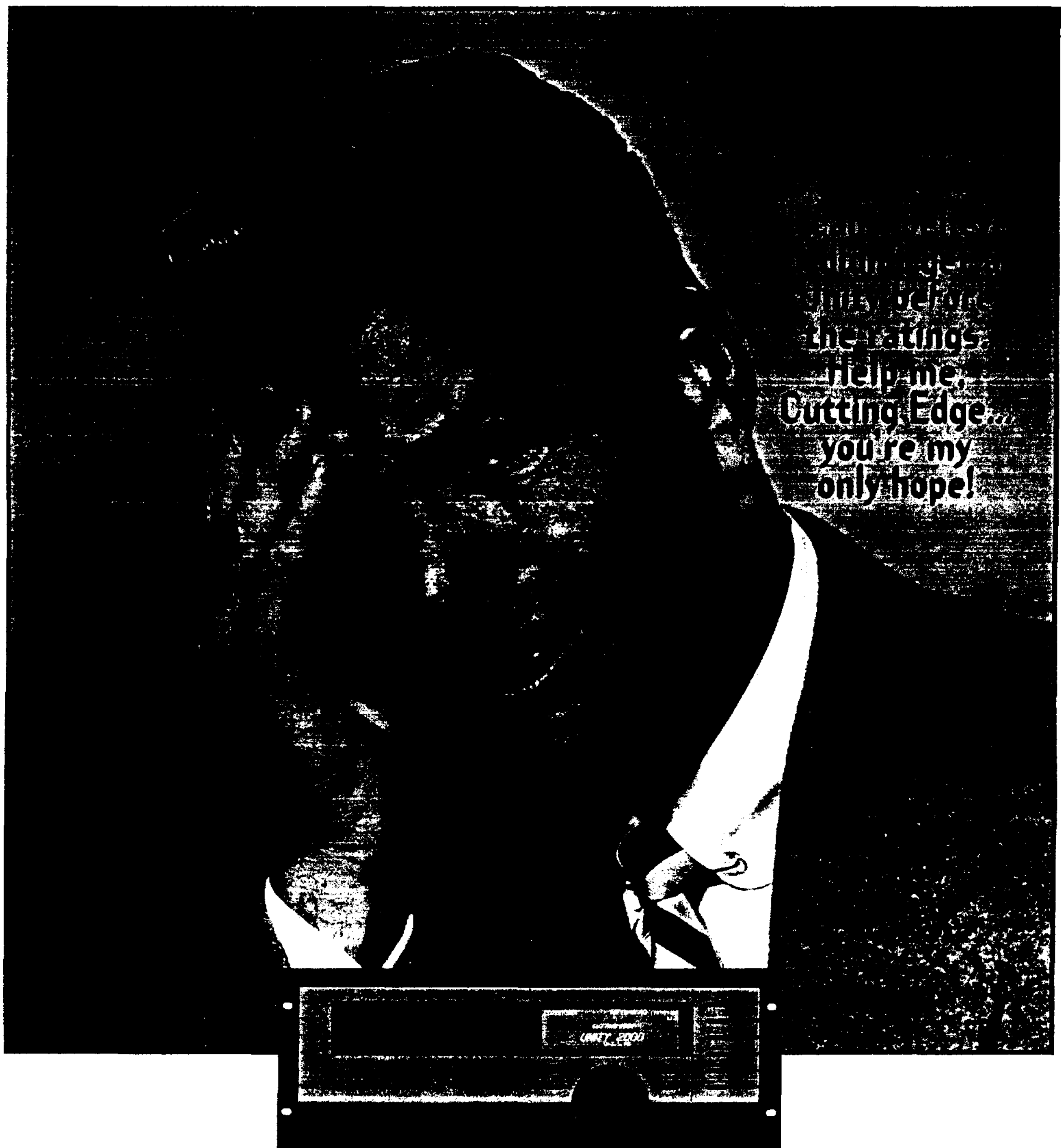
| **I/O** | |
|---|---|
| Input Circuits | Servo Balanced Transformerless |
| Output Circuits | Servo Balanced Transformerless |
| Input Connectors | 3-Pin XLR Female |
| Output Connectors | 3-Pin XLR Male |
| Input Impedance | 19.5k Ohms Unterminated; 600 Ohms by Rear Panel Selectable Terminator- (Lifts in Bypass) |
| Output Impedance | 65 Ohms |
| Input CMRR | Better than 60dB 20 Hz to 20kHz |
| Input RF Rejection | Better than 40dB at 800kHz, Better than 60dB Above 2MHz |

| **MISCELLANEOUS** | |
|---|---|
| Power | 120VAC 50/60Hz 30 Watts (100,220,240 options) |
| Power Fuse | 100/120VAC = 0.375A (SLO); 220/240VAC = 0.25A (SLO) |
| Weight | 5.6 Lbs. (2.54kg) |
| Dimensions | 19" W x 1.75" H x 9.5" D (482.83mm x 44.42mm x 241.12mm) |

*MOL is limited by the peak ceiling setting. The output stage is capable of +25dBu into 600 Ohms.
**dB (PK) = peak value of sinewave.
Aphex Systems is constantly striving to maintain the highest standards. As a result of these efforts, modifications may be made from time to time to existing products without prior notice. Specifications and appearance may differ from those listed or shown.


APHEX SYSTEMS 11068 Randall Street • Sun Valley, CA 91352-2621
(818) 767-2929 • FAX (818) 767-2641


Part No. 02-320-01 Printed in U.S.A. U.S. and Foreign Patents issued and pending.




**unity** **the world's best sounding AM & FM processors.**

**Better sound, more listeners, bigger profits. That's what it's all about.**

CUTTINGEDGE

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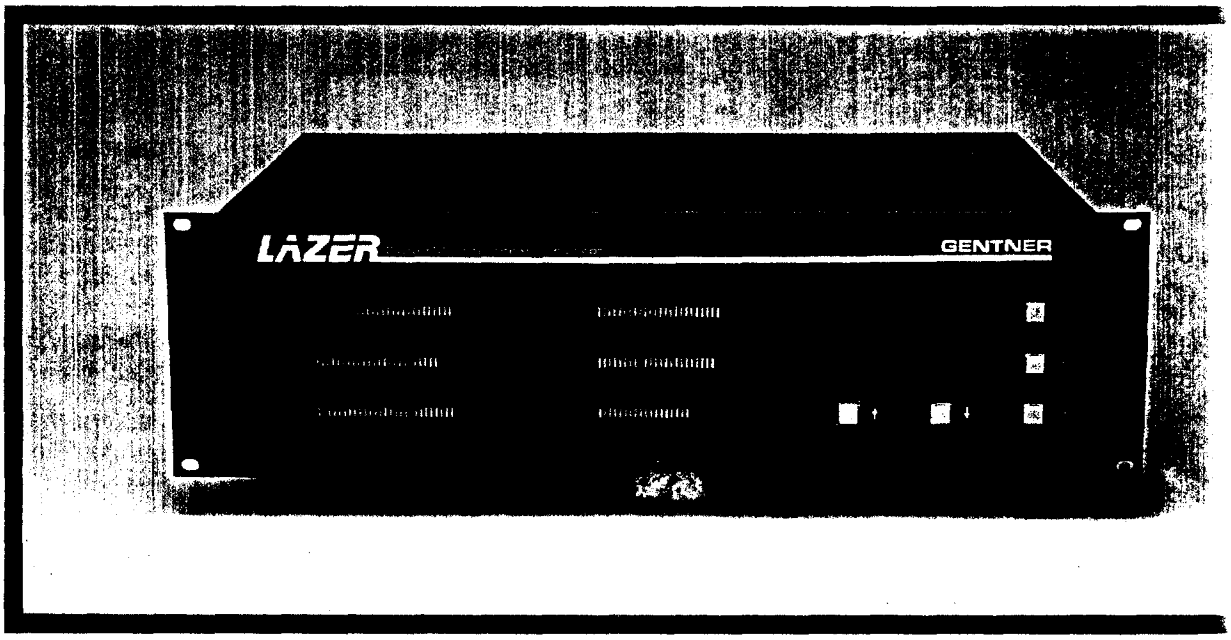
*Preliminary*

# LAZER

*Digital FM Limiter/Stereo Generator*

GENTNER
SPECTRUM
SYSTEMS




**Push button control, digital precision, and total repeatability.**

# Clean, Digital Stereo Generation And Limiting Give You the Competitive Edge.

Digital technology has changed audio quality standards. Audio is cleaner and more transparent than ever. That's the main reason why Compact Discs and digitally recorded audio are so popular with broadcast professionals. On-air broadcasts can sound noticeably better. However, recorded audio, no matter how clear it is, deteriorates through an analog processing chain. Gentner has therefore developed Lazer.

Lazer is the cleanest sounding FM Limiter/Stereo Generator available. Lazer handles audio purely in the digital domain. Because of this, Lazer provides excellent aural specifications including maximum stereo separation, precise limiting, and virtually no noise.

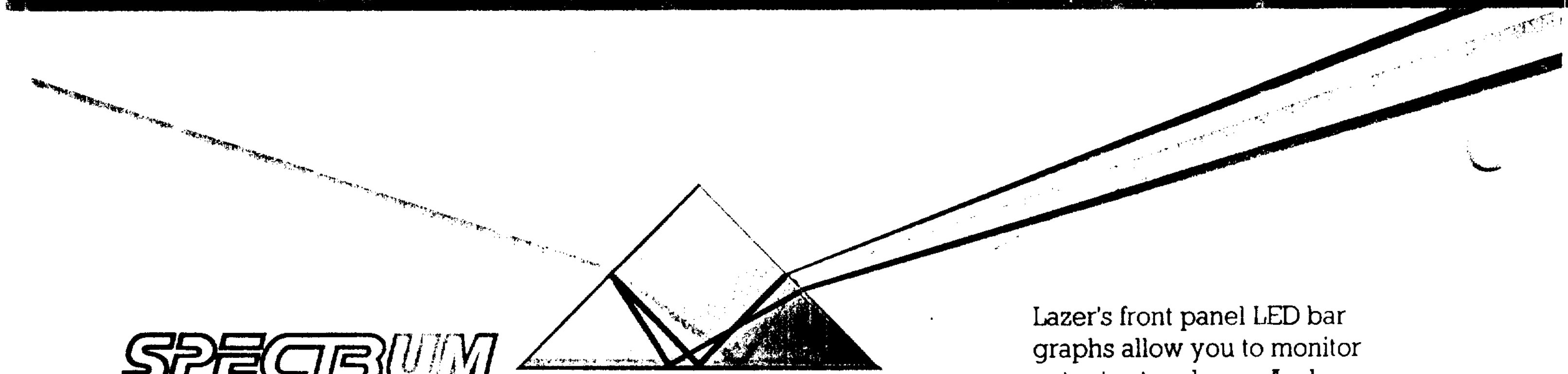
The result: your audio retains its true sound.

Total digital manipulation allows for consistent, precise audio. You receive pure stereo separation and exact limiting. Lazer's 3-band processing is phase linear for maximum transparency. Furthermore, once you've set your parameters they stay set. Tweaking isn't necessary. You can walk away from Lazer and feel confident that your adjustments will never drift.

Front panel buttons and an LCD display menu let you numerically define operation parameters. For example, define pilot injection, release times, limiting threshold, etc. using absolute numerical values. Define these parameters by moving through different display screens and menus. Precise values, along with preset programs, allow you to repeat settings exactly.

With Lazer, there's no limit to your creativity. Lazer provides you with exceptional control. An extensive range of operation parameters, clean manipulation of audio, precise stereo generation with minimal crosstalk, and simple operation let you produce the sound you want. So go ahead and be creative!

Expand your digital audio chain and keep a competitive edge. Try out Lazer and hear for yourself. Its sound will say everything. For more information, give your Lazer/Spectrum System dealer a call today or contact Gentner's sales department at (801) 975-7200.



# SPECTRUM SYSTEMS

Lazer is one component of Spectrum Systems. Spectrum Systems is the broadcast industry's first all-digital on-air processing system. Using Spectrum Systems, you achieve total modulation control and powerful audio processing capabilities. With Lazer, you receive maximum stereo separation and precise limiting while retaining clean, clear audio.

Lazer uses digital signal processing for absolute consistent quality. Digital signal processing provides stability - critical parameters will not change with time or temperature; more accruate control - the use of special algorithms and digital memory provide for precise limiting; and repeatability - because digital processing numerically manipulates audio, you can set exact parameters and repeat those exact parameters when desired.

Lazer also features a high immunity to noise and RFI because its circuitry is all digital. Audio remains transparent.

Lazer is expandable. Lazer's software design allows you to easily upgrade the unit without changing physical components, thereby always keeping you up-to-date with the latest technology.

Maintenance requirements are low. Because of Lazer's digital processing accuracy, periodic recalibration is unnecessary.

**Be creative with your operation parameters.**

Lazer's front panel programming lets you design the sound you want. Operation parameters are numerous and can be specifically defined. For example you can numerically select parameters for individual bands such as release time, attack time, threshold, interband coupling, etc. You can also select exact parameters for the system. Define pilot injection, peak detector threshold, and safety clipper threshold along with other functions. Lazer has its own built-in test signals, accessible by the front panel menu, that make it easy to check out your equipment chain.

Lazer's menu also allows you to use up to eight pre-defined programs. Use the menu to configure a set of parameters, then enter the parameter set as a pre-set program. You can easily and quickly change your processing for different dayparts or to accommodate various formats.

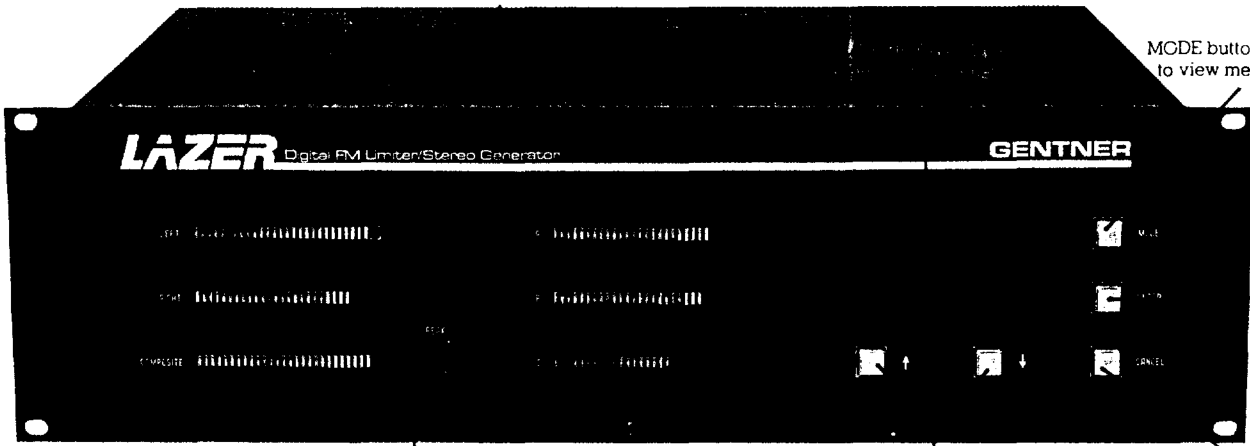
Lazer's front panel LED bar graphs allow you to monitor outputs at a glance. And, you can monitor and control parameters from a remote location. The entire Lazer menu can be connected to a personal computer through the unit's RS-232 port.

**Spectrum Systems - the heart of your audio chain.**

Besides the Lazer FM Limiter/Stereo Generator, Spectrum Systems consists of Prizm, an all-digital 4-band audio processor, and the Optical Encoder, an advanced analog to digital converter. Lazer must be used with the Optical Encoder for conversion of analog audio information into digital data.

*Liquid crystal display shows the status of Lazer's parameters.*

To receive the benefits of an all-digital audio processing system for clear, powerful audio, use the complete Spectrum System: Lazer, Optical Encoder, and the new Digital Prizm multi-band audio processor. Or use Lazer and Optical Encoder with Gentner's legendary Audio Prisms, digitally controlled analog audio processors. Lazer and Optical Encoder's analog inputs and outputs provide for compatibility with your existing equipment.




MODE button allows you to view menu choices

ENTER button allows you to select menu choice.

LED bar graphs provide multi-function readouts at a glance.

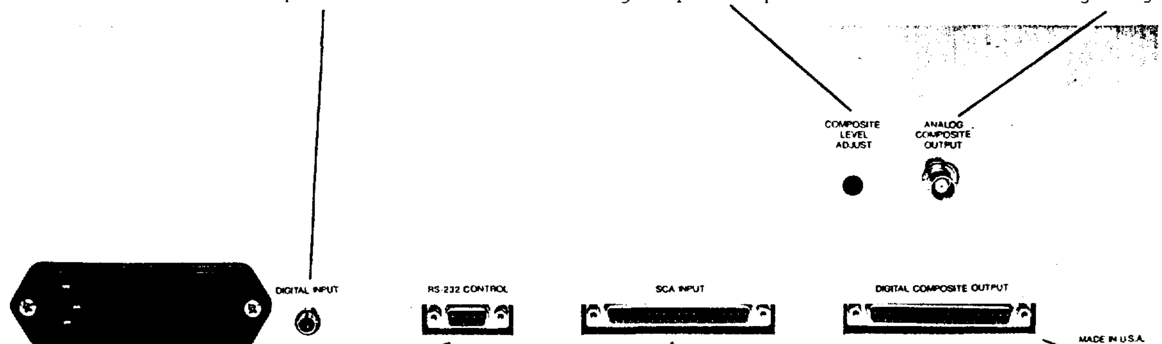
Arrow keys allow you to change operation parameters.

CANCEL button returns you to previous menus.

Data for Left and Right audio channels enter Lazer here from Optical Encoder or Prizm.

Analog Composite Level Adjust. This potentiometer adjusts the amplitude of the analog composite signal available at the Analog Composite Output connector.

Analog Composite Output Connector. The composite analog baseband signal is available here for use with existing analog exciters.




AC power input assembly with fuse.

RS-232 PORT Connector. Can be connected to a modem or PC to provide the same control and display information as the front panel.

Digital SCA Input Connector. 24-bit word input to be used with a digital SCA generator.

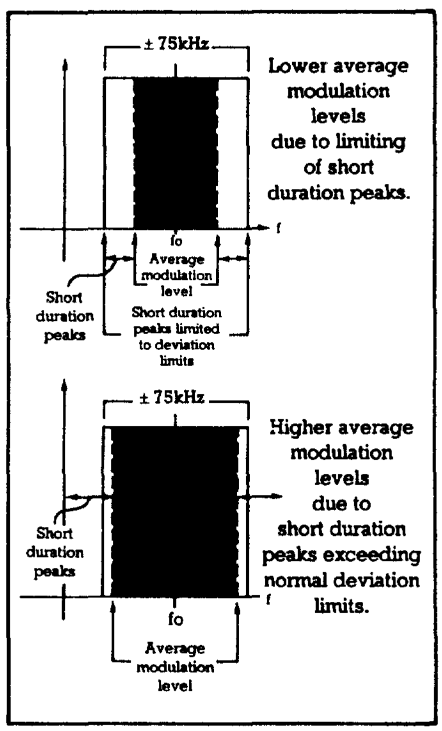
Digital Composite Output Connector. 24-bit word output to be used with a digital FM exciter.

## Advantages of Digital FM Limiting

Lazer's FM limiting is accomplished in the digital domain using numeric processing. This approach provides significant advantages over existing analog limiting. These advantages include: much better brightness and clarity of audio, higher average modulation levels, and cleaner occupied spectrum.

Unlike analog limiters, the digital Lazer uses "smart" processing. This allows Lazer to evaluate signal peaks on the basis of level *and* duration.

The peak indicators on Lazer can be programmed to display two types of peak levels: 1) Instantaneous peak levels. These are the types of peaks that most modulation monitors indicate. 2) Duration qualified peaks. For example Lazer can be programmed to activate its peak indicators only if a peak is longer than one millisecond. Or two milliseconds. And so on. Broadcasters can avoid unnecessary limiting of high frequency material. This results in higher average modulation levels and improved clarity of high frequency material. The diagram at right illustrates the advantage of higher modulation levels when short duration peaks are allowed to go beyond normal deviation limits.



## Preliminary Specifications:

Lazer tested without Optical Encoder unless otherwise specified.

| | |
|---|---|
| **Monophonic signal to noise ratio (with Optical Encoder and 75 microsecond deemphasis):** | 88 dB below 100% modulation |
| **Monophonic signal to noise ratio (75 microsecond deemphasis):** | 100 dB below 100% modulation |
| **Wideband noise (30 kHz low pass filtered):** | -94 dB |
| **Wideband noise (80 kHz low pass filtered):** | -87 dB |
| **Stereo Separation:** | >70 dB @ 400 Hz<br>>65 dB @ 15 kHz |
| **Stereo signal to noise (75 microsecond deemphasis):** | >80 dB below 100% Left or Right modulation |
| **Crosstalk:** | |
| **Main to Sub Channel (with 75 microsecond deemphasis):** | >70 dB |
| **Crosstalk (continued):** | |
| **Sub to Main (with 75 microsecond deemphasis):** | >70 dB |
| **Sub or Main to SCA spectrum (56 kHz to 99 kHz, including stereo pilot harmonics):** | >85 dB |
| **Distortion:** | |
| **Measured at analog composite output, internally generated 1187.5 Hz tone, with 75 microsecond deemphasis):** | <.004% |
| **Distortion: (with Optical Encoder, 75 microsecond deemphasis, L-R @ 1 kHz):** | <.008% |
| **Preemphasis performance:** | ±0.1 dB with respect to 75 microsecond curve of FCC part 73.333 |

All specifications are subject to change without notice.

GENTNER
ELECTRONICS
CORPORATION

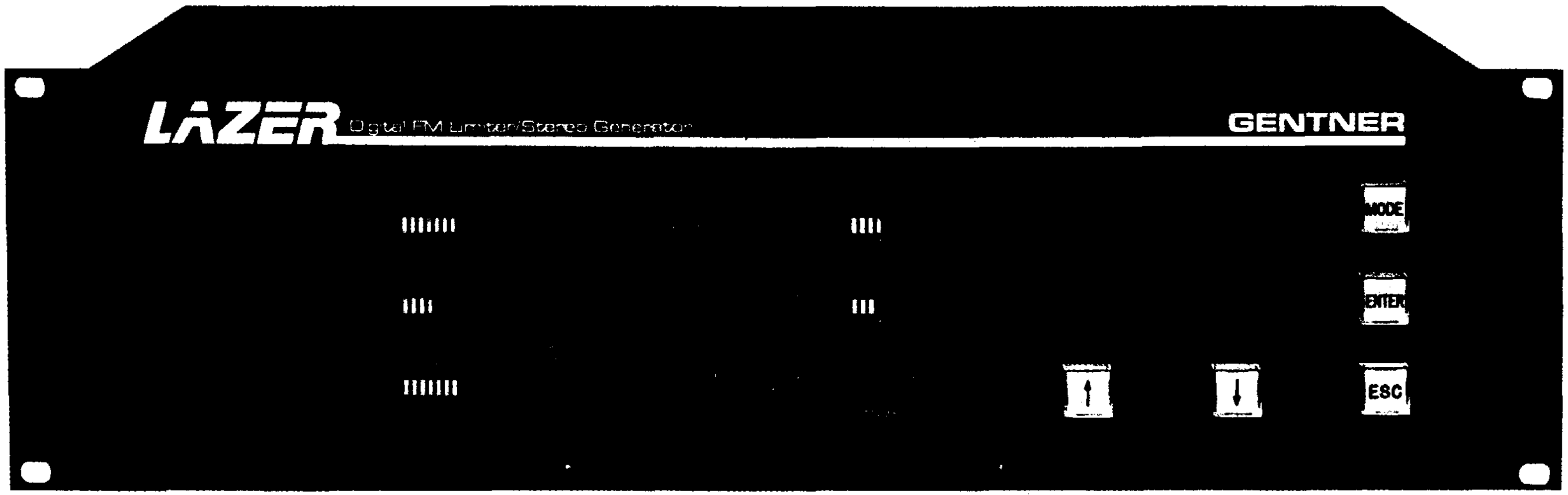
SPECTRUM
SYSTEMS

1825 Research Way
Salt Lake City, Utah 84119-2348
PHONE (801) 975-7200
FAX (801) 977-0087

 2/90

# Introducing the 100% Digital LAZER

## FM Limiter/Stereo Generator




- 100% Digital Processing provides crystal clear audio
- Less distortion, cleaner signal
- Loudness achieved without clipping
- Twenty-three processing parameters
- Ultra precise baseband stereo signal generation
- Simple setup and operation
- Eight preset processing programs, user adjustable
- A/B processing program comparison
- Remotely accessible via RS-232
- Upgradeable through future software programs

# Specifications

## OPTICAL ENCODER:

| | |
|---|---|
| Physical: | 19.0″ X 7.40″ X 1.75″<br>(48.26 cm X 18.8 cm X4.45 cm)<br>6 Lbs. (2.72 kg) |
| Connectors: | Analog input (audio left and right channels): XLR<br>Digital output (stereo): SMA Optical |

## LAZER:

| | |
|---|---|
| Physical: | 19.0″ X 10.60″ X 5.25″<br>(48.26 cm X 26.92 cm X 13.34 cm)<br>15.5 Lbs. (7.04 kg) |
| Connectors: | Digital stereo input: SMA Optical<br>Analog composite baseband output: 75 ohm BNC<br>Digital composite output: DB-37F<br>Digital SCA input: DB-37F<br>RS-232 remote control: DB-9F |

## Wideband Final Limiter:

Provides final limiting without the use of clipping and the resulting distortions.

| | |
|---|---|
| Range of Limiting (Compression) | up to 18 dB |
| Compression Ratio | infinite above an adjustable threshold |
| Attack Time | adjustable from 5 usec to 2 ms |
| Release Time | adjustable from 20 msec to 9.98 seconds |
| Digital Composite Peak Control (DCPC) threshold | adjustable from 75.0% to 133.0% |

## Pre-emphasis:

| | |
|---|---|
| 75 microsecond | Digital Finite Impulse Response Filter accurate to within ±0.1 dB of the FCC curve (part 73.333) |

## Pilot Protection/L-R anti-alias filter:

| | |
|---|---|
| Linear Phase Digital Finite Impulse Response Filter | Passband Ripple DC to 15 kHz ±0.04 dB. Stopband greater than -87 dB at 18 kHz |

## Total Harmonic Distortion:

| | |
|---|---|
| Lazer Alone | < 0.004% with internally generated 1187.5 Hz tone (measured at the analog composite output) |
| Lazer/Optical Encoder | < 0.008% @ 1kHz (analog generator connected to Optical Encoder, measurement made at the analog composite output) |

## FM Limiter Total Harmonic Distortion:

Distortion Measured with 10 dB of limiting in any of the three bands and any combination of tri-band or final limiting totaling 10 dB of compression.

| | |
|---|---|
| Limiter THD @ 10 dB | <0.05% |

## Signal to Noise Ratio:

| | |
|---|---|
| Lazer Alone | >100 dB below full output (133.33% modulation) |
| Lazer/Optical Encoder | >90 dB below full output (133.33% modulation) |

## Stereo Separation:

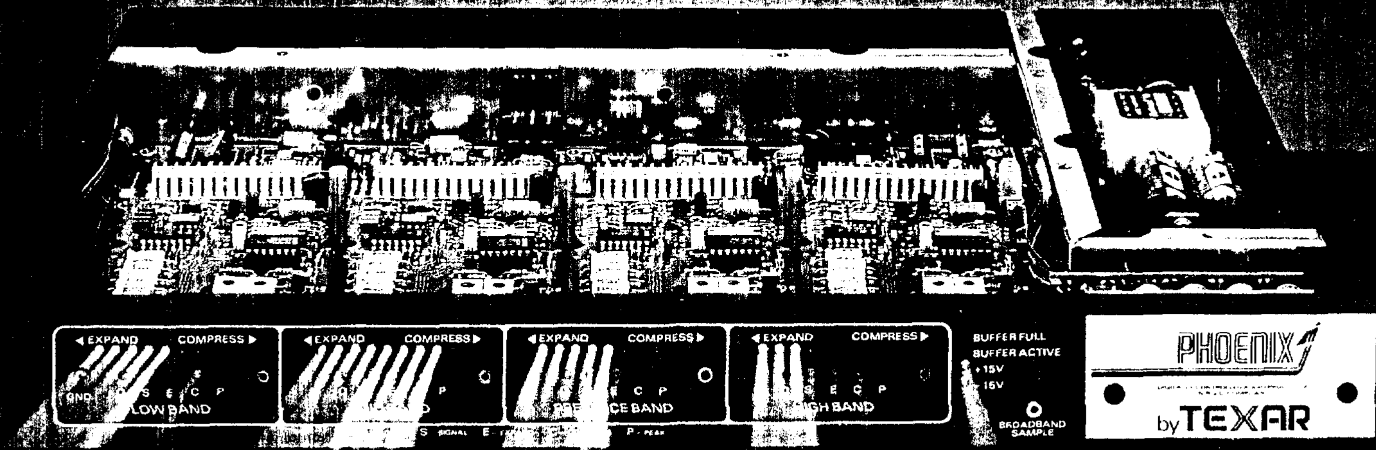
| | |
|---|---|
| Right into Left | >70 dB DC to 400 Hz |
| Left into Right | >65 dB 400 Hz to 15 kHz |

This measurement made at the analog composite output and the measured value usually limited by Modulation Monitor performance.

## Crosstalk:

| | |
|---|---|
| Main into Sub or SCA | >80 dB |
| Sub into Main or SCA | >80 dB |

# HIGH FIDELITY COMES TO AM!




# PHOENIX™

DIGITALLY CONTROLLED AUDIO PROCESSOR
NRSC COMPLIANT

## by TEXAR

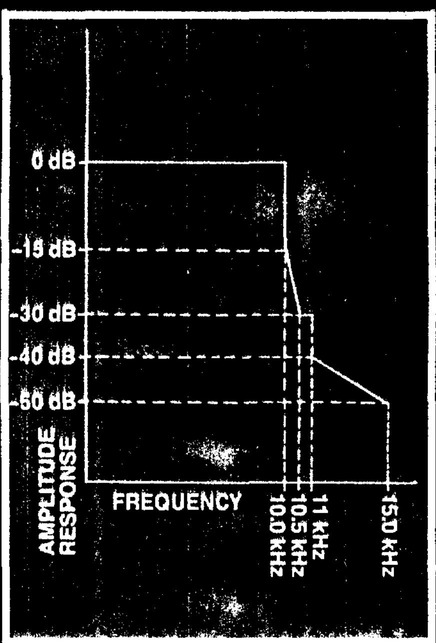
Master

You can buy an NRSC-compliant audio processor from several companies, but only TEXAR can give you NRSC *and* DIGITAL CONTROL. The TEXAR PHOENIX™ packs a clean, powerful wallop in one very small package. For monaural AM, you can't buy more processing power for less money anywhere. Comparing in performance and features with units costing nearly twice as much, the PHOENIX features variable asymmetry, a voice phase-rotator to assure maximum modulation, and a low-frequency tilt-corrector that can compensate for some weaknesses in plate-modulated transmitters.

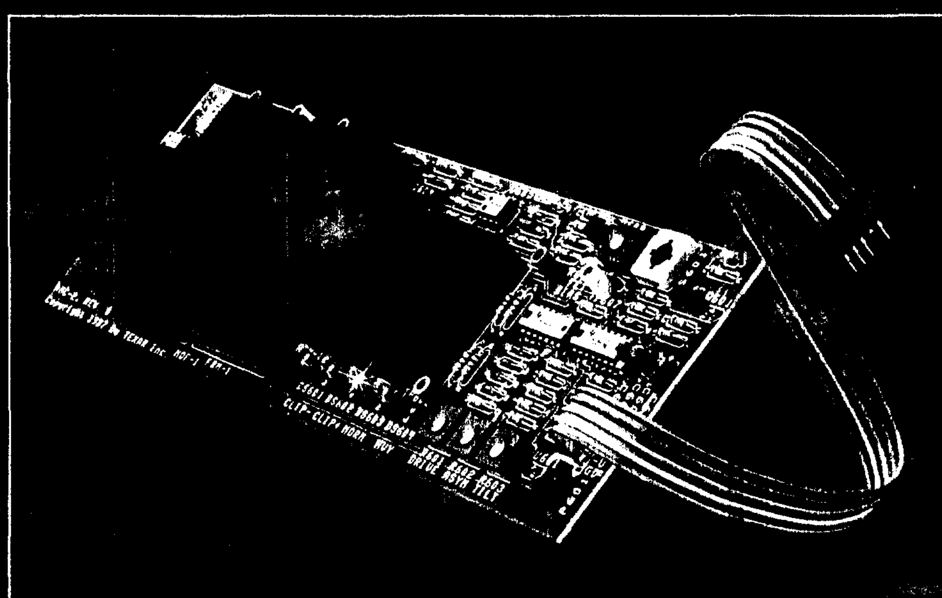
The Phoenix also includes features you don't get with the competition, like the clean, powerful sound of digital control and individual gating of bands so record fades don't "swish up." A simple AC voltmeter (like a Simpson 260 or a Potomac Instruments AA-51) is the only test equipment required for setup. Simply set six voltages to equal the recommended values for your situation. The User's Manual's table of time-proven set-up values insures you get immediate results. The PHOENIX's open-architecture design means repairs or upgrades are only a card-swap away. (PC boards in some other processors are soldered in.)

Proven in the strongest RF fields, the PHOENIX's three-stage RF and transient suppression circuit stops surges and RF interference.

**THE N.R.S.C. 10 kHz LOWPASS FILTER STANDARD**




- **Full NRSC compliance**
- **Advanced digital control produces a clean, powerful signal**
- **Reduces reflected power from narrowband antenna systems**
- **All-in-one construction**
- **Operates in the highest RF environments**
- **Internal phase-rotator**
- **Internal low-frequency tilt corrector with 80 Hz square-wave generator**



*The heart of the Phoenix is the AMC-2 (Amplitude Modulation Controller) circuit card. Previous Texar AM processors can be upgraded to NRSC-spec by replacing the earlier AMC-1 circuit card.*

# PHOENIX

The Phoenix is a complete, monaural, AM, processing system, from console output to transmitter input. It begins with the industry-standard, TEXAR AUDIO PRISM, multi-band, processor. (For additional information, see the three-page AUDIO PRISM brochure.) To this proven performer are added the PR-1 Phase Rotator and the AMC-2 AM Modulation Controller. The NRSC shelving-pre-emphasis and the NRSC 10 kHz "brick wall filter" are implemented on the AMC-2.

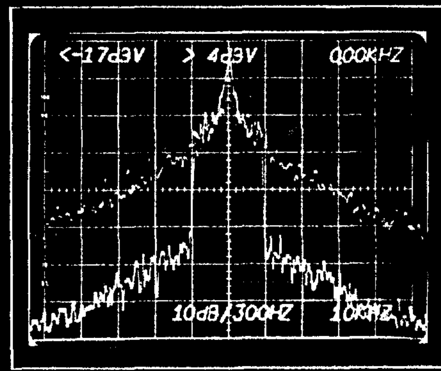
For a copy of the complete NRSC specification, please request the blue TEXAR reprint "INTERIM VOLUNTARY NATIONAL STANDARD." Additional, useful reading is also provided in the yellow TEXAR reprint "THE NRSC STANDARD... WHAT DOES IT MEAN TO YOU? (A non-technical, non-political description of the events leading up to the January 10, 1987 N.R.S.C. standard for pre-emphasis and bandlimiting.)"

The NRSC's pre-emphasis was designed to compensate for the natural high-frequency roll-off of a typical AM radio. As a result, an NRSC transmission will sound normal over most AM radios. However, when received on a broadband system, such as a station's modulation monitor, an NRSC transmission may sound thin or bright. To compensate for this, broadband receiving systems should incorporate a DE-emphasis network that is the exact complement of the transmission, PRE-emphasis curve.

The TEXAR AM RECEPTION FILTER (ARF-1) provides this de-emphasis function in conjunction with a 10 kHz notch filter to eliminate monitor system 10 kHz "whistle" caused by first-adjacent-channel reception.

The ARF-1 also has a 5 kHz, bandlimited output to permit improved reception when night-time, first-adjacent-channel interference is particularly severe. Where optimum off-air reception is obtained with the 10 kHz filter daytime and with the 5 kHz filter nighttime, both filter outputs may be routed to the air console. Where sufficient input positions exist on the console monitor selector switch, both feeds may be made available to the air-staff, thus permitting them to select either wideband or narrowband at will.

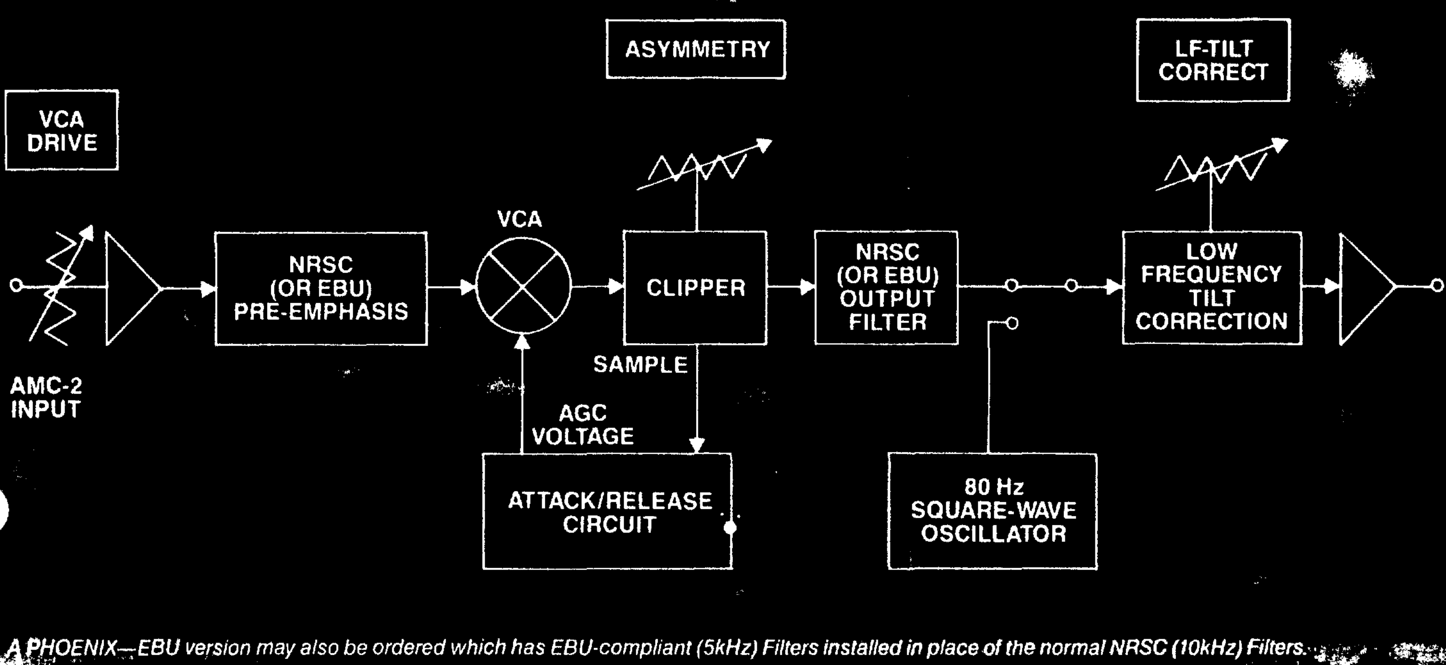
The ARF-1 is a 3" x 5" circuit board having solder-post type input, output and power connections. It is intended to mount inside the station's modulation monitor or monitor receiver. The ARF-1 requires an external supply of bipolar 15 volts for operation. Current demand is low and this power can normally be safely obtained from the modulation monitor's power supply.




*A spectrum-analyzer comparison of the dynamic output of an earlier, wideband, TEXAR AMC-1 (Red line) and the NRSC-compliant TEXAR AMC-2. (Green line)*

*TEXAR engineers used the latest CAD/CAM software to design the PHOENIX.™*

## BLOCK DIAGRAM OF THE TEXAR AMC-2 CIRCUIT CARD



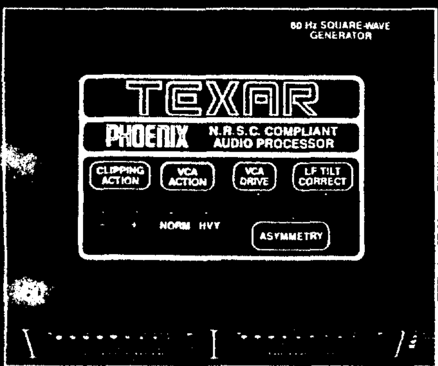

*A PHOENIX—EBU version may also be ordered which has EBU-compliant (5kHz) Filters installed in place of the normal NRSC (10kHz) Filters.*

## NEW LIFE FOR OLD TRANSMITTERS

It goes without saying that the better the transmitter, the better the on-air audio. But sometimes a new transmitter just isn't a possibility. In that case, the PHOENIX can do a lot to pump new performance into an old transmitter.

Many AM transmitters, particularly those employing plate-modulation, exhibit low-frequency tilt. Tilt can rob you of up to 20% of the modulation you're entitled to, having the same effect as reducing the power of a 1 kilowatt station to 640 watts.

The PHOENIX includes a very effective, low-frequency, tilt corrector that "tunes out" this transmitter weakness. (See diagram at right.) Several hundred of these correctors are already in use in the earlier AMC-1. Alignment previously required an ex-




*The AMC-2 includes an internal 80 Hz square-wave oscillator to facilitate adjustment of the low frequency tilt corrector.*

ternal 80 Hz oscillator. The AMC-2 has its own built-in oscillator. Merely press a button and the 80 Hz test signal replaces program audio. Release the button and normal operation is resumed. Output level of the oscillator is deliberately restricted to 60% of peak value to prevent damage to the modulation reactor.

## EASY TO MAINTAIN

The PHOENIX is easy to maintain. Open architecture (plug-in card) construction simplifies troubleshooting. Additionally, in the event of future improvements, you can replace individual cards without obsoleting the entire unit.

To further simplify troubleshooting, 40 color-coded test points allow quick access to key voltages to permit fast fault isolation. Each circuit board also has a silk-screened parts legend to allow you to quickly home in on the desired component.

## OPTIONS

Standard configuration of the PHOENIX includes a rear-panel barrier-strip for audio connections. XLR input/output connectors are available as a standard option where desired.

The PHOENIX is also available in an EBU configuration where EBU-compliant (5 kHz) filters are installed in place of the NRSC (10 kHz) filters. This configuration is useful for international shortwave broadcasting (See FCC Part 73.766), use in radio zones where the standard channel spacing is less than 10 kHz, and where first-adjacent-channel short-spacing makes greater suppression of higher-frequency sidebands desireable. For additional information, please request a copy of the PHOENIX-EBU brochure.

## COMPARE!

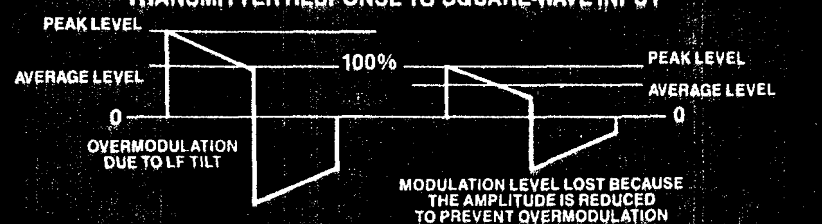
Compare features. Compare performance. Compare ease of customer use. We think you'll choose the TEXAR PHOENIX.

Like most TEXAR products, the PHOENIX is available on a 10-day demo basis to insure that it's right for your needs. Financing is available.

## HOW CORRECTING LOW-FREQUENCY TRANSMITTER TILT INCREASES AVERAGE MODULATION LEVEL.

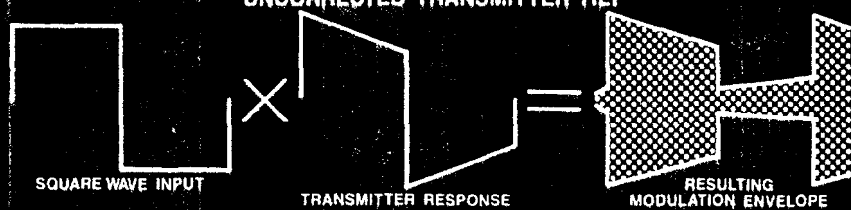
*As shown below, where LF tilt is present, the leading edge of the output square wave has an amplitude substantially higher than that of its average value. As a result, the amplitude of the input signal must be reduced substantially to keep the peak level below 100% and prevent overmodulation. This reduces a station's average modulation.*

**TRANSMITTER RESPONSE TO SQUARE-WAVE INPUT**

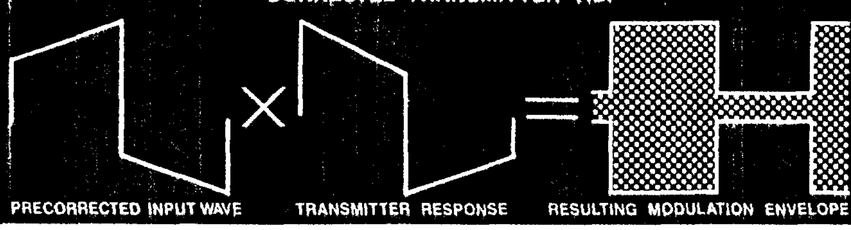



*The Low-Frequency Tilt Correction circuit compensates for this transmitter weakness by precorrecting the signal as shown below. It tilts the plateau of the input wave in the direction opposite that of the transmitter's response. The two wave tilts cancel each other and the transmitter envelope output is level as desired. The signal to the transmitter can then be increased, also increasing the average modulation level.*

**UNCORRECTED TRANSMITTER TILT**




**CORRECTED TRANSMITTER TILT**




TEXAR PHOENIX


A Division of Gentner Electronics Corporati

P.O. Box 27647
Salt Lake City, UT 84127-0647
(801) 975-7200
Telex II 910-380-6909 FAX: (801) 977-0087

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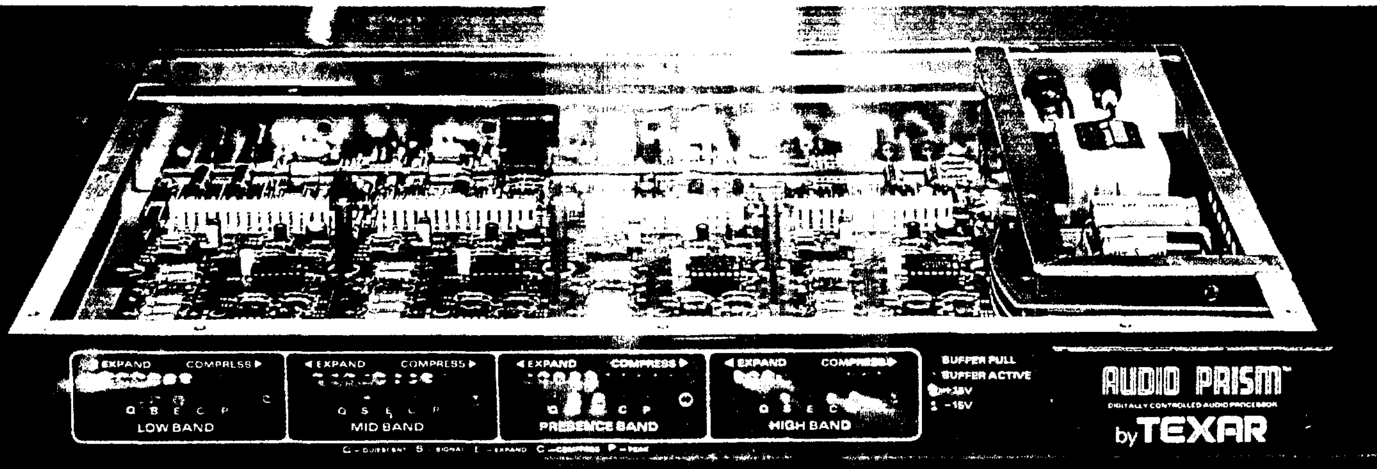
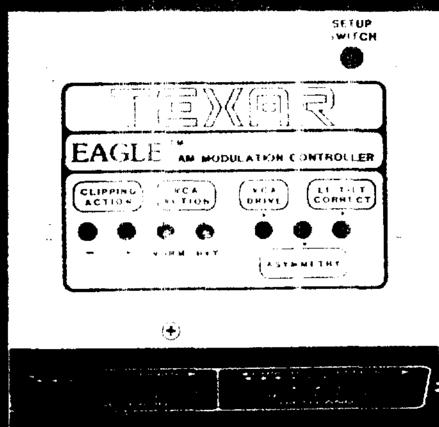



# TEXAR PHOENIX. MAJOR MARKET PERFORMANCE. SMALL MARKET PRICE.

## * N.R.S.C. Compliant * Digital Control * Four-band Power * Only $2,745 List

The only processor to combine digital-control and the National Radio Systems Committee's (NRSC) new voluntary standard for pre-emphasis and bandlimiting, the TEXAR PHOENIX packs a clean, powerful wallop in one, very small package. For monaural AM, you can't buy more processing power for less money anywhere. Comparing in performance and features with units costing nearly *twice as much*, the TEXAR PHOENIX features variable asymmetry, a voice phase-rotator to insure maximum modulation, and a low-frequency tilt-corrector that can compensate for some weaknesses in plate-modulated transmitters.

The PHOENIX also includes features you don't get with the competition, like the clean, powerful sound of digital control and individual gating of bands, so record fades don't "swish up". But high technology doesn't mean complicated installation. A simple AC voltmeter (like a Simpson 260 or a Potomac Instruments AA-51) is the only test equipment required for set-up. Simply set six voltage readings to equal the recommended values for your situation. The Users' Manual's table of time-proven set-up values insures you get immediate results. The PHOENIX's open-architecture design means repairs or upgrades are only a card-swap away. (PC boards in some other processors are soldered in.)




Compare features. Compare performance. Compare ease of customer use. We think you'll choose the TEXAR PHOENIX. At $4,000.00, the PHOENIX would be a good deal. At $2,745.00, it's a steal! Financing is available.

Like all TEXAR products, the PHOENIX is available on a 10 day demo basis to insure that it's right for your needs.



**TEXAR**
**PHOENIX**
**List Price . . . $2,745.00**

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